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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/019,450	05/28/2002	Ravi Chandran	2376.2115-014.	4359

57690 7590 01/16/2009

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EXAMINER	
WOZNIAK, JAMES S	

ART UNIT	PAPER NUMBER
2626	

MAIL DATE	DELIVERY MODE
01/16/2009	PAPER

Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/019,450	CHANDRAN ET AL.	
	Examiner	**Art Unit**	
	JAMES S. WOZNIAK	2626	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *03 November 2008*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-60* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-60* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *14 February 2008* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Response to Amendment

1. In response to the office action from 10/22/2008, the applicant has submitted an a Request for Continued Examination, filed 11/3/2008, amending independent claims 1 and 31, while arguing to traverse the art rejection based on the limitation regarding the gain calculation unit/step and a detection unit/step that detects over-amplification or over-suppression as a function of the calculated gain *(Amendment, Pages 17-18)*. Applicant's arguments have been fully considered, however the previous rejection is maintained due to the reasons listed below in the response to arguments.

2. The IDS filed on 11/3/2008, which includes the reference requested in the Requirement for Information from 7/1/2008, has been considered by the examiner.

3. In response to amended claims 11-24, the examiner has withdrawn the 35 U.S.C. 112, second paragraph rejection directed to indefinite claim language.

Response to Arguments

4. Applicant's arguments have been fully considered but they are not persuasive for the following reasons:

With respect to Claim 1, the applicants argue that the prior art of record does not disclose "a gain calculation unit to calculate a gain value as a function of the first parameter value and the adjusted first parameter value" and "a detection unit to detect over amplification or over suppression of said signal as a function of the gain value" (Amendment, Page 17). The applicants specifically disagree with the examiner's interpretation of Tackin et al *(U.S. Patent: 7,092,365)*, which they argue only teach comparing signals against clipping thresholds to detect "over amplification" and not the aforementioned gain calculation or detection unit *(Amendment, Pages 17-18)*.

In response, the examiner notes that Tackin does teach a gain calculation unit as well as a gain detection means. More specifically, it should first be pointed out that Yajima et al *(U.S. Patent: 5,873,058)* teaches the partial decoding of a speech parameter for gain adjustment *(Col. 9, Line 35- Col. 10, Line 25; Col. 27, Line 40- Col. 29, Line 10; Col. 31, Lines 12-53; and Figs. 13, partial decoder, Element 108 and 17-18 and output of gain adjusted data on a transmission line)*, while Tackin teaches that a speech parameter *(which can be in compressed domain format, Col. 15, Lines 52-53)* is multiplied by a gain factor (i.e., the claimed first parameter adjustment, Fig. 8A, Element 150), a gain value for a subsequent threshold comparison is calculated based on the result of the speech input and the gain adjustment to said input *(i.e., the claimed gain calculation unit relying on a first parameter value and an adjusted first parameter value, Fig. 8A, Element 154)*, and finally a detection unit *(Fig. 8A, Element 156)* subjects the calculated gain value to a threshold comparison for detection of over-amplification *(i.e., the claimed detection unit that detects over-amplification or over-suppression as a function of the calculated gain factor, Col. 22, Lines 36-55)*. Thus, since Tackin does teach the aforementioned claim

limitations, the applicants' arguments have been fully considered, but are not convincing. The applicant traverses the art rejection of claim 31 and the further dependent claims for the same reasons as claim 1 *(Amendment, Pages 18-20)*. In regards to such arguments, see the response directed to claim 1. The applicants' traverse the art rejection of independent claims 25 and 55 and their associated dependent claims for reasons similar to claim 1 *(Amendment, Page 20)*, however these claims do not contain the aforementioned claim limitations. Thus, the applicants' arguments with respect to these claims are moot.

Claim Rejections - 35 USC § 112

5. The following is a quotation of the first paragraph of 35 U.S.C. 112:

> The specification shall contain a written description of the invention, and of the manner and process of making and using it, in such full, clear, concise, and exact terms as to enable any person skilled in the art to which it pertains, or with which it is most nearly connected, to make and use the same and shall set forth the best mode contemplated by the inventor of carrying out his invention.

6. **Claims 1-24 and 31-54** are rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the written description requirement. The claim(s) contains subject matter which was not described in the specification in such a way as to reasonably convey to one skilled in the relevant art that the inventor(s), at the time the application was filed, had possession of the claimed invention.

Claims 1 and 31 *(and also their associated dependent claims)* feature a claim limitation that calculates a gain value based on an adjusted first parameter and a first parameter. It appears that, in the specification *(Page 27, for example)* the gain calculator could be determining the described desired gain, but its calculation does not rely on an

input speech signal and an adjusted gain value, thus the added gain calculation means/step fails to comply with the written description requirement.

7. The following is a quotation of the second paragraph of 35 U.S.C. 112:

> The specification shall conclude with one or more claims particularly pointing out and distinctly claiming the subject matter which the applicant regards as his invention.

8. **Claims 1-24 and 31-54** are rejected under 35 U.S.C. 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention.

Claims 1 and 31 *(and also their associated dependent claims)* feature a claim limitation that calculates a gain value based on an adjusted first parameter and a first parameter, but it is uncertain which component this step/mean refers to in the specification. The applicants' specification features several variations of a gain control circuit, but it appears that the claimed invention could refer to ALC loop shown/described in Fig. 13. In this embodiment, a realized gain is multiplied by an original speech level *(i.e., the "second generator" unit/step that adjusts a first characteristic, Specification, Page 27, Lines 10-20)* and compared to the determined desired gain. It appears that the gain calculator could be this desired gain, but its calculation does not rely on an input speech signal and an adjusted gain value, thus it is unclear to what unit/step the gain calculator is referring. The claim will be interpreted as it is set forth for the application of the prior art of record.

Claim Rejections - 35 USC § 103

9. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

10. **Claims 1-6, 10-11, 15, 24, 31-36, 38, 40-41, 45, and 54** are rejected under 35 U.S.C. 103(a) as being unpatentable over Jarvinen et al *(U.S. Patent: 5,946,651)* in view of Yajima et al *(U.S. Patent: 5,873,058)* and further in view of Tackin et al *(U.S. Patent: 7,092,365)*.

With respect to **Claims 1 and 31**, Jarvinen discloses:

In a communications system for transmitting digital signals using a compression code comprising a predetermined plurality of parameters including a first parameter, said parameters representing an audio signal comprising a plurality of audio characteristics including a first characteristic, said first parameter being related to said first characteristic *(receiving transmitted coded speech parameters at a decoder including LPC coefficients and a gain parameter, Col. 6, Lines 16-58)* said compression code being decodable by a plurality of decoding steps including a first decoding step for decoding said parameters related to said first characteristic, an apparatus for adjusting the first characteristic comprising:

A decoder responsive to said digital signals to read at least said first parameter and to A generator for generating at least a first parameter value derived from said first

parameter *(means for decoding and generating excitation parameters having associated gain factors, Col. 6, Lines 16-58)*;

Responsive to said digital signals and said first parameter value a second generator to generate an adjusted first parameter value representing an adjustment of said first characteristic *(adjusting a gain factor with a scaling factor, Col. 7, Line 58- Col. 8, Line 61)*; and

Responsive to said adjusted first parameter value a replacement unit to derive an adjusted first parameter and to replace said first parameter with said adjusted first parameter *(means for replacing an excitation parameter and associated gain with a perceptually adjusted excitation parameter, Col. 7, Line 34- Col. 8, Line 61)*.

Jarvinen does not teach the concept of adjusting a gain parameter over a network prior to reception at a receiver, however Yajima discloses the concept of speech signal gain parameter adjustment at a relay node *(Col. 9, Line 35- Col. 10, Line 25; Col. 27, Line 40- Col. 29, Line 10; Col. 31, Lines 12-53; and Figs. 13, partial decoder, Element 108 and 17-18 and output of gain adjusted data on a transmission line)*.

Jarvinen and Yajima are analogous art because they are from a similar field of endeavor in speech decoding utilizing adaptive gain control. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Jarvinen with the concept of gain adjustment at a relay node as taught by Yajima in order to implement gain adjustment at a means that is capable of connecting two different types of networks *(Col. 3, Lines 25-27)* and coding voice efficiently *(Col. 7, Lines 25-32)*, while also inherently decreasing the amount of processing performed at a receiver.

Although Jarvinen teaches the concept of gain control and Yajima teaches that gain control can be performed in the compressed domain, Jarvinen and Yajima fail to explicitly teach a further gain adjustment step wherein a detection of over or under amplification is performed and a feedback loop is utilized. Tackin, however, recites the gain-adjusting a speech signal, calculating a gain value based on the gain-adjusted input speech signal, and comparing that value to a threshold means for detecting over/under amplification and the use of a feedback loop in further automatic gain control *(Fig. 8A; and Col. 22, Line 7- Col. 23, Line 15; and input signal in the compressed domain, Col. 15, Lines 52-53).*

Jarvinen, Yajima, and Tackin are analogous art because they are from a similar field of endeavor in speech processing utilizing adaptive gain control. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Jarvinen in view of Yajima with the AGC concept taught by Tackin in order to ensure that voice signals are maintained at an acceptable volume *(Tackin, Col. 22, Lines 8-10).*

With respect to **Claims 2 and 32**, Jarvinen discloses:

The first characteristic comprises a level of the audio signal *(gain factor that is indicative of a desired speech signal level, Col. 5, Line 25- Col. 6, Line 32; and Col. 12, Lines 24-33).*

With respect to **Claims 3 and 33**, Yajima further discloses:

Yajima teaches avoiding synthesizing filter processing for a normal voice signal that would not require gain adjustment *(Col. 22, Lines 1-24; and adjusting a gain speech*

parameter, Col. 9, Line 35- Col. 10, Line 25; Col. 27, Line 40- Col. 29, Line 10; Col. 31, Lines 12-53).

With respect to **Claims 4 and 34**, Yajima teaches synthesizing filter processing, as applied to Claims 3 and 33.

With respect to **Claims 5 and 35**, Jarvinen discloses:

The compression code comprises a linear predictive code *(LP coefficients, Col. 5, Lines 25-57).*

With respect to **Claims 6 and 36**, Jarvinen discloses:

The compression code comprises regular pulse excitation long term prediction code *(LTP prediction coefficients, Col. 5, Lines 25-57).*

With respect to **Claims 10 and 40**, Yajima further discloses gain adjustment implementation at a relay node situated on a network that would inherently be capable of receiving near and far end speech from various transmission nodes connected to the network *(Fig. 16, Element 404; Col. 9, Line 35- Col. 10, Line 25; Col. 27, Line 40- Col. 29, Line 10; Col. 31, Lines 12-53).*

With respect to **Claims 11 and 41**, Jarvinen discloses:

The processor test the adjusted first parameter value for an overflow and underflow condition before deriving the adjusted first parameter *(multiple threshold comparisons, Col. 7, Line 58- Col. 8, Line 61).*

With respect to **Claims 15 and 45**, Jarvinen discloses performing the decoding processing, as applied to Claim 1, on a plurality of parameters from a series of time frames *(Col. 6, Lines 16-58; and Col. 12, Lines 52-54).*

With respect to **Claims 24 and 54**, Jarvinen further discloses:

The processor performs at least the first decoding step to generate decoded signals related to the first characteristic of the audio signal *(recovering speech parameters using a speech decoder, Col. 6, Lines 16-26)*.

With respect to **Claim 38**, Jarvinen discloses the use of the CELP coding standard *(Col. 5, Lines 25-35)*.

11. **Claims 8-9, 12, 16, 18, 20-23, 39, 42, 46, 48, and 50-53** are rejected under 35 U.S.C. 103(a) as being unpatentable over Jarvinen et al in view of Yajima, in view of Tackin et al and further in view of Yasunaga et al *(U.S. Patent: 6,330,534)*.

With respect to **Claim 8**, Jarvinen in view of Yajima and further in view of Tackin discloses the speech decoding apparatus utilizing perceptual gain scaling, as applied to Claim 1. Jarvinen in view of Yajima and further in view of Tackin does not explicitly teach the use of the algebraic CELP coding standard, however Yasunaga teaches the use of said standard *(Col. 3, Lines 42-51)*.

Jarvinen, Yajima, Tackin, and Yasunaga are analogous art because they are from a similar field of endeavor in speech decoding utilizing adaptive gain control. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Jarvinen in view of Yajima with the ACELP standard taught by Yasunaga in order to provide a standard that reduces the complexities of computing coding distortions *(Yasunaga, Col. 3, Lines 42-51)*.

With respect to **Claim 9**, Jarvinen further teaches the gain scaling factor as applied to claim 1.

With respect to **Claims 12 and 42**, Jarvinen in view of Yajima and further in view of Tackin discloses the speech decoding apparatus utilizing perceptual gain scaling, as applied to Claims 11 and 41. Jarvinen in view of Yajima and further in view of Tackin does not teach that a decoder derives an adjusted speech parameter by quantizing an adjusted speech parameter, however Yasunaga discloses a process for adjusting a gain factor applied to a speech parameter by quantizing an adjusted target speech parameter *(Col. 30, Line 42- Col. 31, Line 9)*.

Jarvinen, Yajima, Tackin, and Yasunaga are analogous art because they are from a similar field of endeavor in speech decoding utilizing adaptive gain control. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Jarvinen in view of Yajima with the gain adjusting process taught by Yasunaga in order to provide a means for minimizing a quantization error between target and decoded speech parameters *(Yasunaga, Col. 30, Line 42 - Col. 31, Line 9)*.

Claims 16 and 46 contain subject matter similar to Claims 12, 15, 42, and 45 and thus, are rejected for the same reasons.

Claims 18 and 48 contains subject matter similar to Claim 12, and thus, is rejected for the same reasons.

With respect to **Claims 20 and 50**, Yasunaga further discloses scalar quantization performed using a predetermined quantization table *(Col. 12, Lines 10-21)*.

With respect to **Claims 21 and 51**, Yasunaga further discloses subframe-based speech processing *(Col. 1, Line 33- Col. 2, Line 9)*.

With respect to **Claims 22 and 52**, Yasunaga further discloses:

The processor replaces the first parameter with the adjusted first parameter for a first subframe before processing a subframe following the first subframe *(adjusting gains of processing frames within a speech frame on a frame-by-frame basis, Col. 28, Lines 40-50; and Col. 30, Line 42- Col. 31, Line 9).*

With respect to **Claims 23 and 53**, Yasunaga further discloses adjusting a gain of a current processing frame based on a gain of a previous processing frame *(Col. 30, Line 42- Col. 31, Line 9)*, and subframe-based speech processing, as applied to Claims 21 and 51.

Claim 39 contains subject matter similar to Claims 9 and 21, and thus, is rejected for the same reasons.

12. **Claims 7 and 37** are rejected under 35 U.S.C. 103(a) as being unpatentable over Jarvinen et al in view of Yajima et al in view of in view of Tackin et al in view of Yasunaga et al *(U.S. Patent: 6,330,534)* and further in view of Crouse et al *(U.S. Patent: 4,899,384).*

With respect to **Claims 7 and 37**, Jarvinen et al in view of Yajima in view of Tackin and further in view of Yasunaga teaches the speech decoding apparatus utilizing gain scaling, subframe based processing, and quantization processing, as applied to Claims 6, 21, 36, and 51. Jarvinen et al in view of Yajima in view of Tackin and further in view of Yasunaga does not specifically suggest utilizing a maximum absolute value of a speech parameter to derive a speech scaling factor, however Crouse teaches the use of such a value *(Col. 5, Lines 5-16).*

Jarvinen, Yajima, Tackin, Yasunaga, and Crouse are analogous art because they are from a similar field of endeavor in speech coding systems. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Jarvinen et al in view of Yajima in view of Tackin and further in view of Yasunaga with the maximum absolute value parameter taught by Crouse in order to implement a speech coded method having reduced peak information that is consistent with a desired speech output quality *(Crouse, Col. 4, Lines 1-11)*.

13. **Claims 13-14, 17, 19, 43-44, 47, and 49** are rejected under 35 U.S.C. 103(a) as being unpatentable over Jarvinen et al in view of Yajima et al in view of Tackin et al in view of Yasunaga et al, and further in view of Swaminathan et al *(U.S. Patent: 5,751,903)*.

With respect to **Claims 13, 17, 19, 43, 47, and 49,** Jarvinen et al in view of Yajima in view of Tackin and further in view of Yasunaga teaches the speech decoding apparatus utilizing perceptual gain scaling and quantization processing, as applied to Claims 12, 16, 18, 42, 46, and 48. Jarvinen et al in view of Yajima in view of Tackin and further in view of Yasunaga does not teach the use of differential scalar quantization, however Swaminathan discloses the use of such a quantization during speech coding *(Col. 10, Lines 48-56)*.

Jarvinen, Yajima, Tackin, Yasunaga, and Swaminathan are analogous art because they are from a similar field of endeavor in speech coding systems. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Jarvinen et al in view of Yajima in view of Tackin and further in view of

Yasunaga with the differential scalar quantization taught by Swaminathan in order to implement a means for quantizing speech parameters that requires a reduced number of bits *(Swaminathan, Col. 8, Lines 65-98)*.

With respect to **Claims 14 and 44**, Yasunaga further discloses the use of a feedback loop having a speech parameter quantizer *(Col. 30, Line 42- Col. 31, Line 9; and Fig. 16)*, while Swaminathan discloses the use of differential scalar quantization as applied to Claims 13 and 33.

14. **Claims 25-30 and 55-60** are rejected under 35 U.S.C. 103(a) as being unpatentable over Yajima et al *(U.S. Patent: 5,873,058)* in view of in view of the Applicants' Admitted Prior Art *(AAPA)*.

With respect to **Claims 25 and 55**, Yajima discloses:

A transmitter transmitting digital signals using a compression code comprising a predetermined plurality of parameters including a first parameter, said parameters representing an audio signal comprising a plurality of audio characteristics including a first characteristic, said first parameter being related to said first characteristic *(transmission node that outputs a coded voice signal, Col. 16, lines 52-60; wherein voice parameters comprise CELP coded speech and associated gain data, Col. 1, Line 33- Col. 2, Line 7; and Col. 9, Lines 52-57)* wherein said compression code being decodable by a plurality of decoding steps including a first decoding step for decoding said parameters related to said first characteristic *(decodable speech parameters including a step for extracting voice parameters from a voice code signal, Col. 9, Line 35- Col. 10, Line 25; Col. 21, Lines 39-50)*; and

A processor responsive to said second bits to adjust said first bits and said second bits, whereby said first characteristic is adjusted *(adjusting a gain speech parameter at a relay device, Col. 9, Line 35- Col. 10, Line 25; Col. 27, Line 40- Col. 29, Line 10; Col. 31, Lines 12-53)*,

Wherein the processor adjusts the first characteristic without decoding said compression code *(partial decoding of speech parameters, Col. 33, Lines 42-65)*.

A transmitter to transmit digital signal with adjusted first bits and second bits to a device to produce a corresponding audible signal with the first characteristic in the adjusted state *(Fig. 13)*.

Yajima does not explicitly recite the combination of a compression code and a linear code to express a speech signal, however, such a coding scheme is well known in the prior art as is evidenced by the AAPA. The AAPA recites a TFO GSM standard using a combination of coded speech and PCM bits *("TFO standard", Page 25, Line 21- Page 26, Line 4)*.

Yajima and the AAPA are analogous art because they are from a similar field of endeavor in speech compression. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Yajima with the TFO GSM standard recited in the AAPA in order allow Yajima's gain controller to comply with well-known cellular network standards *(AAPA, Page 25, Lines 21-22)*.

With respect to **Claims 26 and 56**, AAPA recites:

The linear code comprises PCM code *(PCM samples, Page 26, Line 4)*.

With respect to **Claims 27 and 57**, Yajima discloses:

The first characteristic comprises audio level *(gain parameter which is indicative of an audio level, Col. 9, Line 35- Col. 10, Line 25; Col. 27, Line 40- Col. 29, Line 10; Col. 31, Lines 12-53)*.

With respect to **Claims 28 and 58**, the AAPA recites the TFO GSM standard as applied to Claims 26 and 57.

With respect to **Claims 29 and 59**, the AAPA further recites first bits comprising the two LSBs and second bits comprising 6 MSBs *(Page 26, Lines 2-3)*.

With respect to **Claims 30 and 60**, the AAPA further recites the use of PCM code for the 6 MSBs *(Page 26, Lines 2-3)*.

Conclusion

15. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure: See PTO-892.

16. Any inquiry concerning this communication or earlier communications from the examiner should be directed to James S. Wozniak whose telephone number is (571) 272-7632. The examiner can normally be reached on M-Th, 7:30-5:00, F, 7:30-4, Off Alternate Fridays.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Patrick Edouard can be reached at (571) 272-7603. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

/James S. Wozniak/
Primary Examiner, Art Unit 2626